IN THE SPECIFICATION

At page 4, first paragraph, lines 9 to 10 and line 13:

Most speech coding systems in use today are based on telephone-bandwidth narrowband speech, nominally limited to about 200-3400 Hz and sampled at a rate of 8 kHz. The inherent bandwidth limitations cause degradation to the communication quality. Recently, there are various efforts to develop wideband speech (band-limited to about 20 ~ 7000 Hz) coding systems surpassing the quality of conventional telephone-bandwidth speech. The 3rd Generation Partnership Project (3GPP) and the International Telecommunication Union-Telecommunication (ITU-T) have recognized the importance of wideband speech and had selected the Adaptive Multi Rate - WideBand (AMR-WB), a.k.a. and ITU-T G.722.2 as their wideband speech codec standard. And also the 3rd Generation Partnership Project 2 (3GPP2) goes through with its own wideband speech codec standard. Thus narrowband speech network and wideband speech codec standard. Thus narrowband speech ~~network~~ networks and wideband speech ~~network~~ networks may co-exist in the near future. When networks employing the different codec standard are inter-networking through the gateway system, there is a need for translation of the coded bit ~~steam~~stream. Generally, when we interlink the networks employing the different codecs with the different bandwidths, we need more sophisticated translation skill. This translation operation is so called (trans-coding." The conventional and simple solution is that an encoder part of one codec is concatenated to a decoder part of the other codec.

At page 8, fourth paragraph, lines 17 to 18:

The formant parameter translator 32 translates ~~a~~ formant parameters encoded in an input CELP format into an output CELP format and generates formant parameters in the output CELP format.